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BANDWIDTH COMPRESSION FOR HDTV BROADCASTING:

Investigation of some adaptive subsampling strategies

M.J. Knee, M.A., N.D. Wells, B.A., D.Phil.

Research Department, Engineering Division
THE BRITISH BROADCASTING CORPORATION

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M.J. Knee, M.A., N.D. Wells, B.A., D.Phil.

Summary

Two possible adaptive subsampling strategies for bandwidth compression of an HDTV signal are described and the results of computer-based investigations of their performance are given. The strategies, known as 'slope' coding and 'BASS' (Block Adaptive Sub-Sampling) coding, are examples of 'revolutionary' or non-compatible versions of DATV (Digitally Assisted TeleVision). The results indicate that either of the strategies could be used to achieve a fourfold compression in bandwidth, enabling the transmission of an HDTV signal within a single WARC 77 DBS channel, but that BASS coding is the more promising of the two in terms of output picture quality, particularly when it is used in conjunction with motion compensation. Some practical considerations relating to hardware design are also discussed.

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M.E.B. Moffat

Head of Research Department

Research Department, Engineering Division,
BRITISH BROADCASTING CORPORATION

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1. INTRODUCTION

The BBC, among others, is investigating various methods of reducing the bandwidth of signals which may be used in a future High Definition Television (HDTV) broadcasting system. At present such a system of broadcasting is unspecified, but experimenters are fully aware that considerable bandwidth reduction will be required if HDTV signals are to be sent using either existing transmission channels or proposed new DBS channels.

For the purpose of this Report, it is assumed that the bandwidth of the production studio HDTV signal will be about 50 MHz for the luminance and colour-difference signals combined. One goal for a broadcast HDTV system is to enable such a signal to be carried by a single WARC 77 DBS channel in the 11.7 to 12.5 GHz band. These channels have a nominal bandwidth of 27 MHz for frequency-modulated signals, and will probably accommodate a maximum baseband bandwidth of about 12 MHz. Therefore, if HDTV broadcasting within a DBS channel is to be achieved, the bandwidth of a studio generated signal must be reduced by a factor of at least four.

One bandwidth reduction system that gives a 4-to-1 compression is MUSE[1]. This system divides the picture into 'still' (or uniformly panned) areas and 'moving' areas on a block-by-block basis, the status of each block being detected at the decoder from the transmitted signal. The blocks are subsampled uniformly for transmission, but the pre- and post-filtering associated with the subsampling is different for the two modes, so that more spatial bandwidth is preserved for still areas and more temporal bandwidth for moving areas. The MUSE system does not produce perfect output pictures; in particular, uniformly moving parts of some pictures might not be transmitted with sufficient spatial bandwidth, and the sudden switching between modes which can occur in non-uniformly moving areas can also be somewhat disturbing.

Various studies have been carried out at BBC Research Department on methods of achieving the necessary compression, of which MUSE may be regarded as a simple example. A concept known as Digitally Assisted Television (DATV) has evolved, in which an additional, high-rate digital signal (the digital assistance) is transmitted. This digital signal tells the receiver decoder how to reconstruct the picture from the highly compressed transmitted analogue signal. For example, the digital assistance data might carry motion vector information to the decoder. One advantage of DATV over other techniques is that the need for complex decision-making is restricted to the coder in which, moreover, subsequent improvements can be implemented without affecting the design of the decoders.

One possible requirement of an HDTV transmission system is that the bandwidth-reduced signal should be 'compatible' with future 625-line MAC receivers. This can be arranged to some extent for approaches which use essentially regular, fixed sampling structures. Such systems are termed 'evolutionary' and one possible evolutionary system is described in Ref. 2. If the requirement for compatibility is dropped, however, the sampling structure can be made irregular so that transmitted analogue samples can be concentrated in areas of the picture where they are most needed. The auxiliary digital signal is then used to tell the decoder where in the picture the transmitted samples have come from. Such an 'incompatible' or 'revolutionary' approach, based on adaptive subsampling, is the subject of this Report.

Two methods of bandwidth reduction are described here, termed respectively 'slope' coding and BASS (Block Adaptive Sub-Sampling) coding. In slope coding, a decision is made on a sample-by-sample basis as to whether the sample should be transmitted or whether it can be reconstructed by linear interpolation between nearby transmitted samples. In BASS coding, the picture is split up uniformly into rectangular blocks and a decision is made on a block-by-block basis as to whether the block is transmitted to full resolution or whether it is to be reconstructed from information in the previous frame.

Experiments for the two methods described were carried out using computer simulation on a VAX 11/750 computer, connected to semiconductor sequence stores capable of displaying, for example, four full-size *RGB* frames or 48 quarter-size monochrome frames.

2. ADAPTIVE SUBSAMPLING

Both methods described use adaptive subsampling, i.e. active and inactive areas of the input

picture signal are treated differently. Fig. 1 is a block diagram of an adaptive subsampling coder. For simplicity, only the luminance component is shown. The input signal is converted into digital form at a sampling rate which is high enough to give a full-resolution picture. The adaptive subsampling is then carried out on this digital signal. This process gives rise to the 'digital assistance' or auxiliary data signal which, possibly in a bit-rate reduced form, is transmitted to the decoder. The samples resulting from the adaptive subsampling process will emerge at an irregular rate. They are therefore written into a buffer store so that they can be read out at a regular rate for conversion back into analogue form for transmission. A feedback control path is normally necessary between the buffer store and the subsampling process to prevent buffer overflow or underflow; however, an alternative approach to this problem is described in Section 4.5.

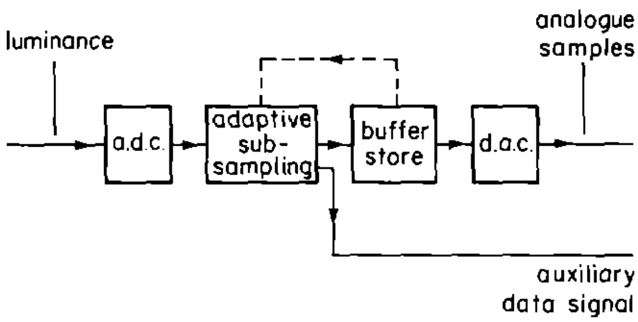


Fig. 1 - Block diagram of adaptive subsampling coder.

It is expected that the data signal would be sent in a burst, possibly at the beginning of each line, field or picture. The capacity of a DBS channel, in terms of the number of independent samples per second that can be sent in analogue form or in terms of the instantaneous data rate that can be supported using advanced forms of modulation, has not yet been accurately defined. If we assume that the maximum possible basebandwidth for a DBS channel is 12 MHz, then the channel should be capable of carrying 24 Msamples per second if used exclusively for carrying the samples in analogue form. When a proportion of the transmission time is occupied by digital data it is assumed that the instantaneous data rate of the digital signal is 48 Mbit/s. This estimate for the available digital bit rate has deliberately been kept lower than what is achievable because it is recognized that there would have to be a substantial overhead for error protection.

3. SLOPE CODING

3.1 Principle of slope coding

In the simplest form of slope coding, the coder selects a proportion of samples from the input signal to transmit to the decoder. The decoder reconstructs the missing samples by linear interpolation between the transmitted samples. Fig. 2(a) shows how the samples near a rising edge might be reconstructed; it should be noted that the straight lines in the diagram are used merely to illustrate the interpolation process and do not show the exact shape of the decoded analogue waveform. This basic technique is not new; the approach has been described with application to video coding by Davisson[3] and Ehrman[4] as 'fan interpolation', and also by Limb[5] and Netravali[6]. However, the method described in this Report differs from these in several details.

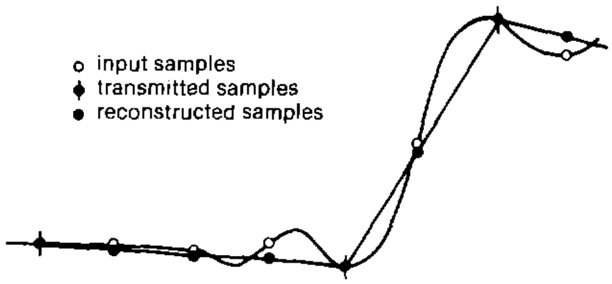


Fig. 2(a) - Principle of slope coding.

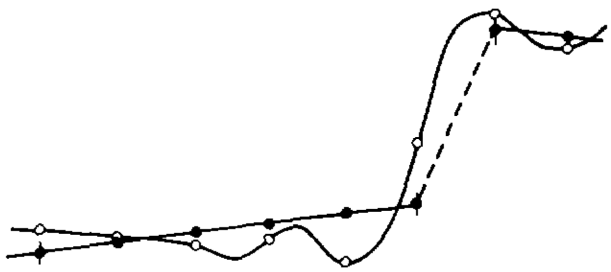


Fig. 2(b) - Improved slope coding.

The coder selects the transmitted samples on the basis of a simple threshold fidelity criterion, in which no decoded sample is permitted to differ from the corresponding input sample by more than a certain threshold, T. To achieve this, while minimizing the number of transmitted samples, the following procedure is adopted: each new input sample is considered in turn as a potential transmitted sample. If the resulting interpolated signal, between the previously transmitted sample and the current sample, fails to meet the threshold criterion, the sample immediately before the current sample is transmitted and the process begins again. The value of T is controlled by feedback from the coder's buffer store.

Using this technique, high quality coding can be achieved on still pictures with an average sample-rate reduction factor of about 4. However, the technique as described is not very suitable for coding noisy source pictures because the subsampling process visibly increases the low-frequency content of the noise spectrum. Inaccuracies in transmitted source samples,

which may have been due to high-frequency noise on the source picture, are transformed into inaccuracies of similar size which propagate across most of the interpolated line. To reduce the effect of this problem, the use of an improved version of slope coding has been adopted which will now be described.

3.2 Improved slope coding

In this version, the decoded samples are derived from a minimum-mean-square-error best-fit straight line approximating the corresponding input samples, as illustrated in Fig. 2(b). The transmitted samples are now the end points of each best-fit line segment. It follows that they occur in pairs of adjacent samples, although the system could be arranged to transmit single samples in extremely detailed areas of the picture.

3.3 Improvements to the threshold criterion

The average transmitted sample rate for a given subjective picture quality can be reduced by about an additional 12% by increasing the value of the threshold T according to some measure of picture activity and also according to the mean brightness level. This can be done because quantization noise is less visible in active areas and also in dark or bright areas of the picture. An effective and readily available measure of activity is the magnitude of the interpolated slope. Fig. 3 shows how, in our experiments, the value of T depended on the slope magnitude and mean brightness level.

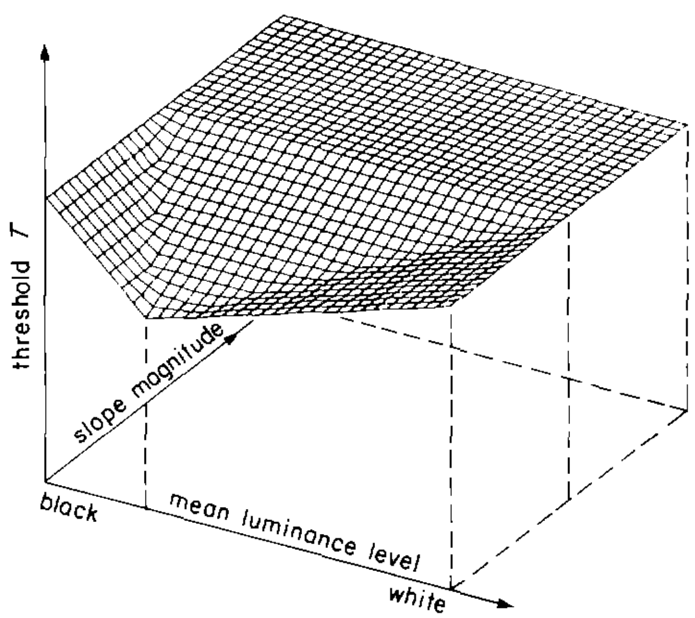


Fig. 3 - Dependence of threshold upon mean level and slope.

3.4 Signalling rate

The most severe apparent disadvantage of slope coding is that the digital data rate required to indicate the positions of the transmitted samples (the auxiliary data) is rather high. Initially, the data rate is equivalent to one bit per input sample to signal whether each given sample value is transmitted or is to be interpolated. If this rate were not reduced in some way, the digital data signal would occupy the whole of the transmission channel! Obviously, therefore, some form of data rate reduction must be used.

One way to reduce the data rate would be to encode the data signal using a variable-length code such as a Huffman code[7]. In practice, this would have to be done on groups of data bits. If no use is made of correlation between data bits (apart from the fact that transmitted samples occur in pairs) and an average proportion p of input samples are transmitted, then a lower bound on the signalling rate per input sample, which is derived in Appendix 1, is:

$$H = -(1 - p/2) \,.\, [q \log_2 q + (1 - q)\log_2(1 - q)]$$

where $q = \dfrac{p/2}{1 - p/2}$.

This lower bound is plotted in Fig. 4 as a function of p.

An alternative way of reducing the data rate, which gives similar results to the above, is to encode the run lengths of line segments using a variable-length code. However, a lower theoretical signalling rate can be obtained if the line-to-line correlation of run lengths is exploited. Fig. 4 also shows this lower bound, which is the measured entropy of run lengths conditional on the run length in the previous television line directly above the first sample in the current line segment in the same field.

The two curves given in Fig. 4 are useful in providing lower bounds for the signalling rate at two

levels of complexity. However, in the simulation of a slope coding system, a simpler coding method was used for the data signal. This was a fixed-length code for the run lengths of line segments. The length of this code needs to be chosen carefully, as it limits the maximum run length and determines the average data rate. For example, if a 4-bit code is used, the maximum run length is 16. At low values of p, limiting the maximum run length can significantly increase the number of transmitted samples for a given picture quality. It was found that, for values of p around $p = 1/6$ and by making the assumptions about transmission of colour difference signals given below, a code length of 5 bits was optimum. This gives a signalling rate of 5 bits for 12 input samples on average, which is equivalent to 0.42 bits per input sample.

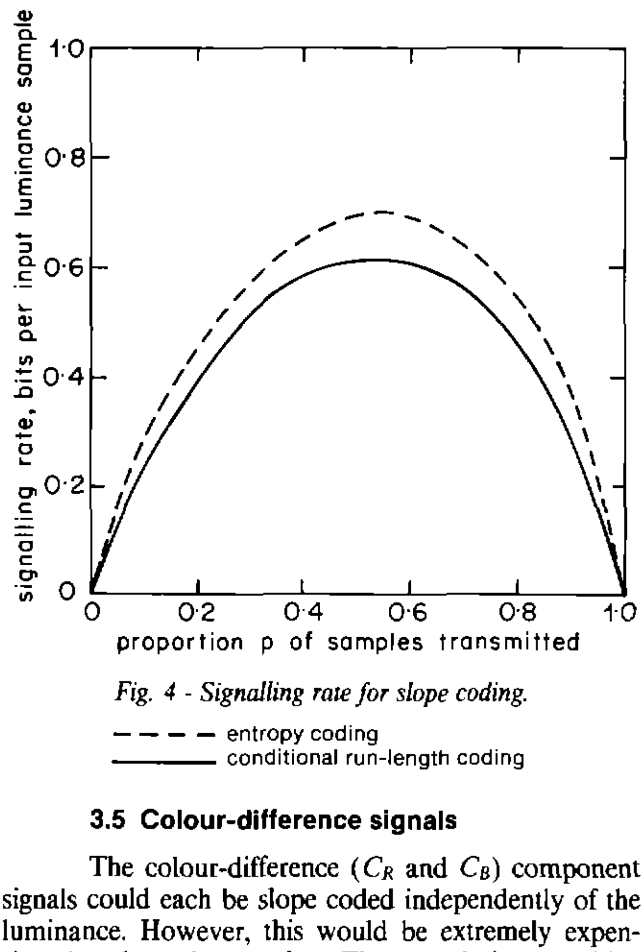


Fig. 4 - Signalling rate for slope coding.

– – – – entropy coding
——— conditional run-length coding

3.5 Colour-difference signals

The colour-difference (C_R and C_B) component signals could each be slope coded independently of the luminance. However, this would be extremely expensive in channel capacity. Three techniques which together reduce the channel capacity occupied by the colour-difference signals have been investigated in some preliminary experiments:

(1) The C_R and C_B signals were each vertically filtered, 2:1 vertically subsampled and transmitted on alternate field-lines.

(2) In general, the positions of transmitted C_R or C_B samples along a line coincide with the positions of transmitted luminance samples. The same data signal can therefore be used for both luminance and colour-difference signals. In order not to miss occasional colour-difference transitions which are not accompanied by significant luminance transitions, the threshold criterion for sample selection must be modified to include the colour-difference signals. The penalty in terms of unnecessary transmitted luminance samples was found to be very low.

(3) The preliminary experiments indicate that forcing the colour-difference slope to zero, and therefore transmitting only one colour-difference sample for each line segment, gives acceptable results.

Taking these three techniques together, we may assume that the colour-difference signals will add nothing to the digital data rate and will contribute an analogue component which requires a channel capacity equal to half that of the analogue part of the luminance component.

3.6 Slope coding for a DBS channel

Bearing the above considerations in mind, we can postulate a 'package' in which each line segment is described by two luminance samples, one colour-difference sample and 5 bits of data. In order to transmit this package in a DBS channel a maximum proportion $p = 0.17$ of input luminance samples may be transmitted. Fig. 5 gives the results of a computer simulation in which this value of p was used, together with an indication of the positions of the transmitted samples. The source picture is a quarter of one frame of the 'Voiture'* sequence. The picture is 360 pixels by 288 lines and therefore represents about 1/20 of the area of a full HDTV picture.

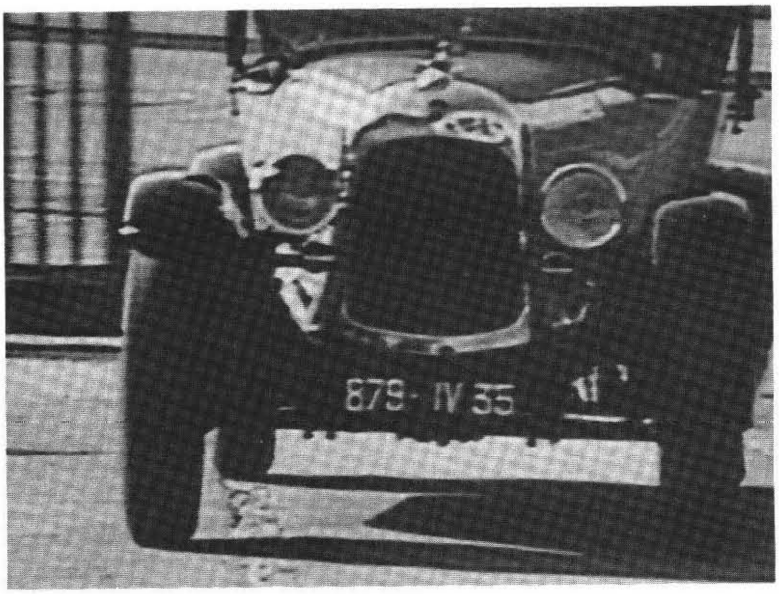
By comparing the result of the simulation, Fig. 5(b), with the source picture, Fig. 5(a), it should be possible to see some impairments caused by the slope coding process, particularly at the left-hand edge of the radiator and in the dark areas around the edges of the tyres, and also by the loss of the vertical lines at the front edges of the doors. However, high-contrast detail such as the moving gate, the shadows and the number plate are reconstructed almost perfectly.

3.7 Possible improvements

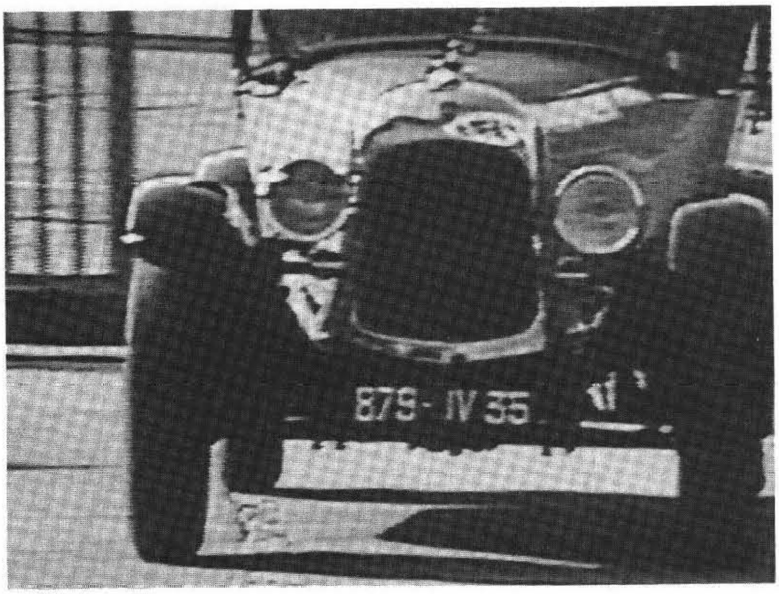
Some extensions of the slope coding principle have been tried; for example, use of the basic

* The Voiture sequence data was kindly provided by CCETT, Rennes, France.

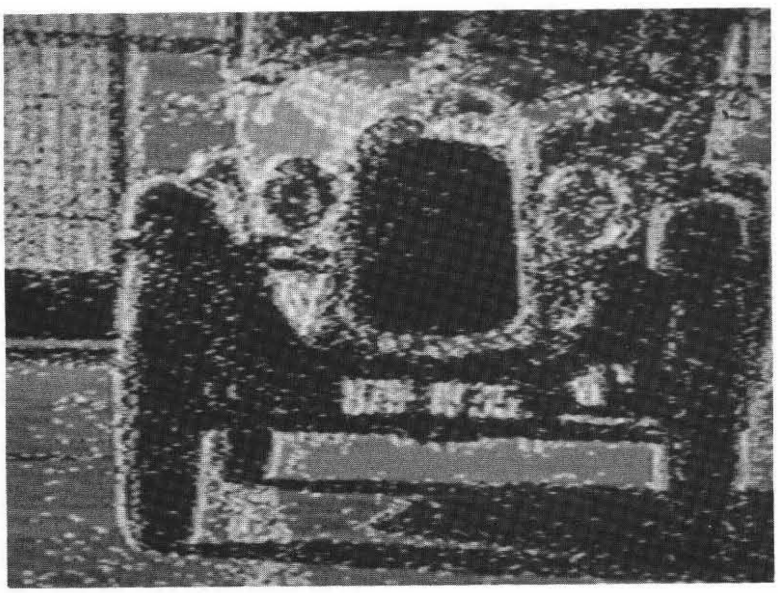
Fig. 5 - Slope coding results.



(a) One frame of the 'Voiture' source sequence.



(b) Picture after slope coding and reconstruction.



(c) Slope-coded picture with transmitted samples highlighted.

technique in the vertical or temporal dimension. Very similar results, in terms of the acceptable sample rate reduction factor, are obtained in all three dimensions.

An improvement was obtained by splitting each picture into rectangular blocks of varying shapes and sizes and applying the same kind of interpolation techniques and threshold criteria in two dimensions, but it is thought that such a coder would be excessively complicated.

3.8 Practical considerations

A block diagram of a possible implementation is shown in Fig. 6. One of the most complicated parts of the coder is the generator of the best-fit slope parameters (Fig. 6(b)). A recursive algorithm and possible implementation for best-fit parameter generation is given in Appendix 2. There are also problems of processing speed connected with the recursive nature of the coding algorithm itself. A degree of parallel processing, for example on an alternate-line basis, could be used to overcome these problems.

The decoder is much simpler than the coder. There is no loop and no need to calculate the best-fit slope parameters.

In order to control the average sample rate and associated data rate it is necessary to incorporate some form of feedback from the transmission buffer to the slope coder. The average transmission rate is varied by varying the threshold function T described by Fig. 3 according to the coder buffer occupancy.

Buffer stores are also required at the decoder. These receive samples and data at a regular rate from the transmission channel and deliver information at the irregular rate required by the slope decoder. It will be necessary to incorporate buffer strategies to ensure that the system works well in the presence of transmission errors in the digital data signal, using techniques such as those described in Ref. 8.

4. BASS CODING

4.1 Principle of BASS coding

The principle of BASS coding is to decide, on a block-by-block basis, between two or more modes of subsampling, each of which may involve a different number of transmitted samples. In the work described here, blocks are typically 4 x 4 or 8 x 8 and there are two subsampling modes: 'high resolution' in which all or nearly all of the samples in the block are transmitted, and 'low resolution' in which only a very few samples (or even none at all) are transmitted. The choice of subsampling mode for each block has to be transmitted to the decoder as a digital data signal. This arrangement overcomes the major problem of the slope coding principle, that of the high data rate needed, because only one bit of digital data is required for each block.

In order to decide whether to transmit a block at high or low resolution the 'activity' of the block is measured. If the activity exceeds some threshold, the block, known as an 'active' block, is transmitted at high resolution. 'Inactive' blocks are transmitted at low resolution. Possible activity measures will be discussed below. The decision threshold must, of course, be allowed to vary in order to limit the average transmitted sample rate.

The experiments described below involved only the luminance component of the picture. However, when calculating how many blocks can be transmitted at high resolution, it has been assumed that the luminance component would occupy three-quarters of the total satellite channel with the colour-difference components together occupying the remaining quarter because of their lower horizontal and vertical bandwidth requirements.

It should be noted that, in the experiments, the blocks were taken from adjacent picture lines rather than field lines.

4.2 Spatial BASS coding

Initial experiments on BASS coding used a spatial activity measure, for example:

$$A = \sum_{i} (x_i - \overline{x})^2$$

where x_i are the samples in the block and $\overline{x}$ is the mean of those samples.

Active blocks (those for which A is greater than a threshold T) were transmitted at high resolution, whereas for each inactive block ($A < T$) only one sample, corresponding to $\overline{x}$, was transmitted. In the decoder, samples in inactive blocks were reconstructed by linear interpolation between the means of surrounding blocks.

The picture quality resulting from such a coding method, using 4 x 4 blocks, was found to be very poor, indicating, as might be expected, that BASS coding can only be used if there is a temporal element in the algorithm.

4.3 Spatiotemporal BASS coding

In its simplest form, this technique is the same as conditional replenishment (described in Ref. 8)

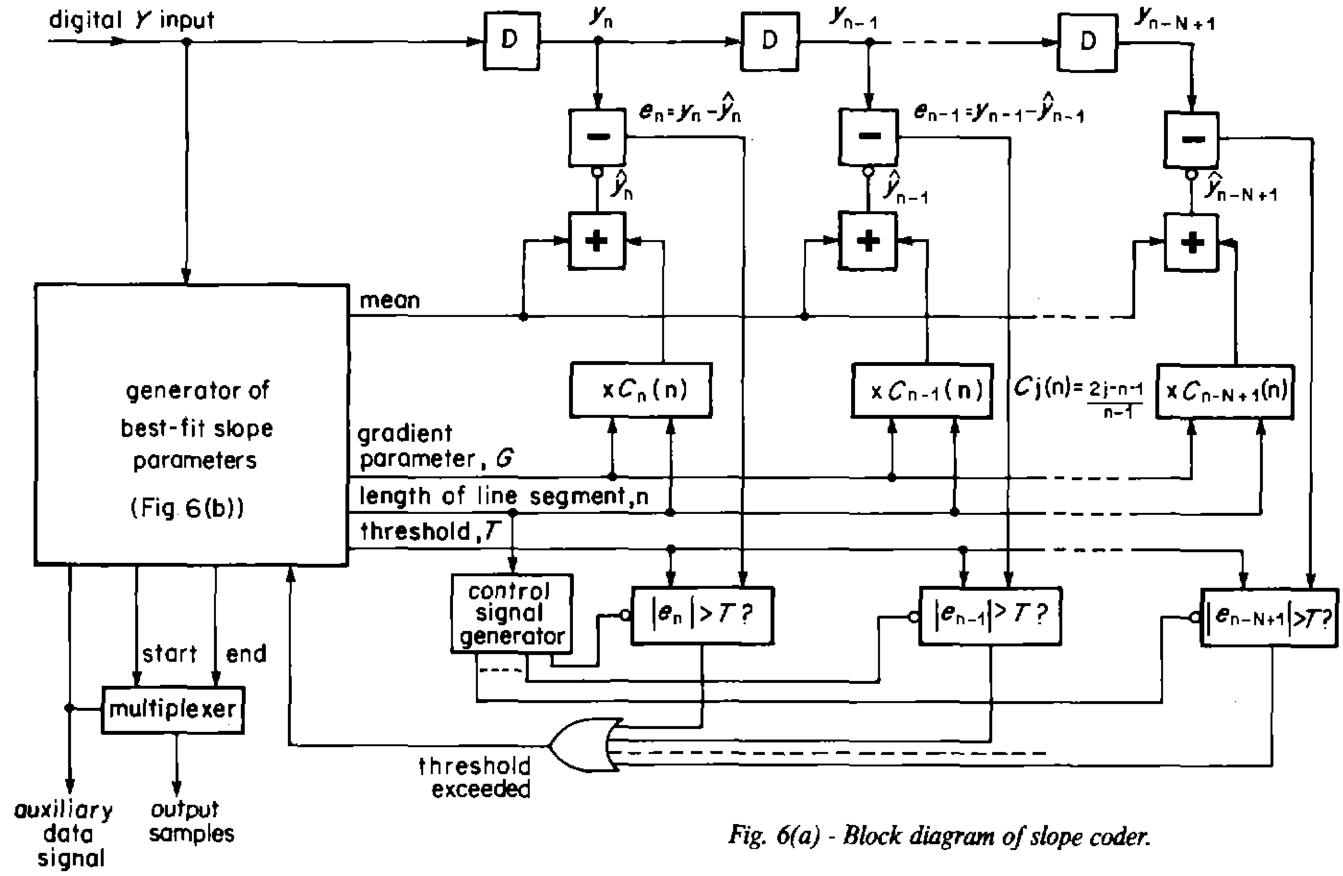


Fig. 6(a) - Block diagram of slope coder.

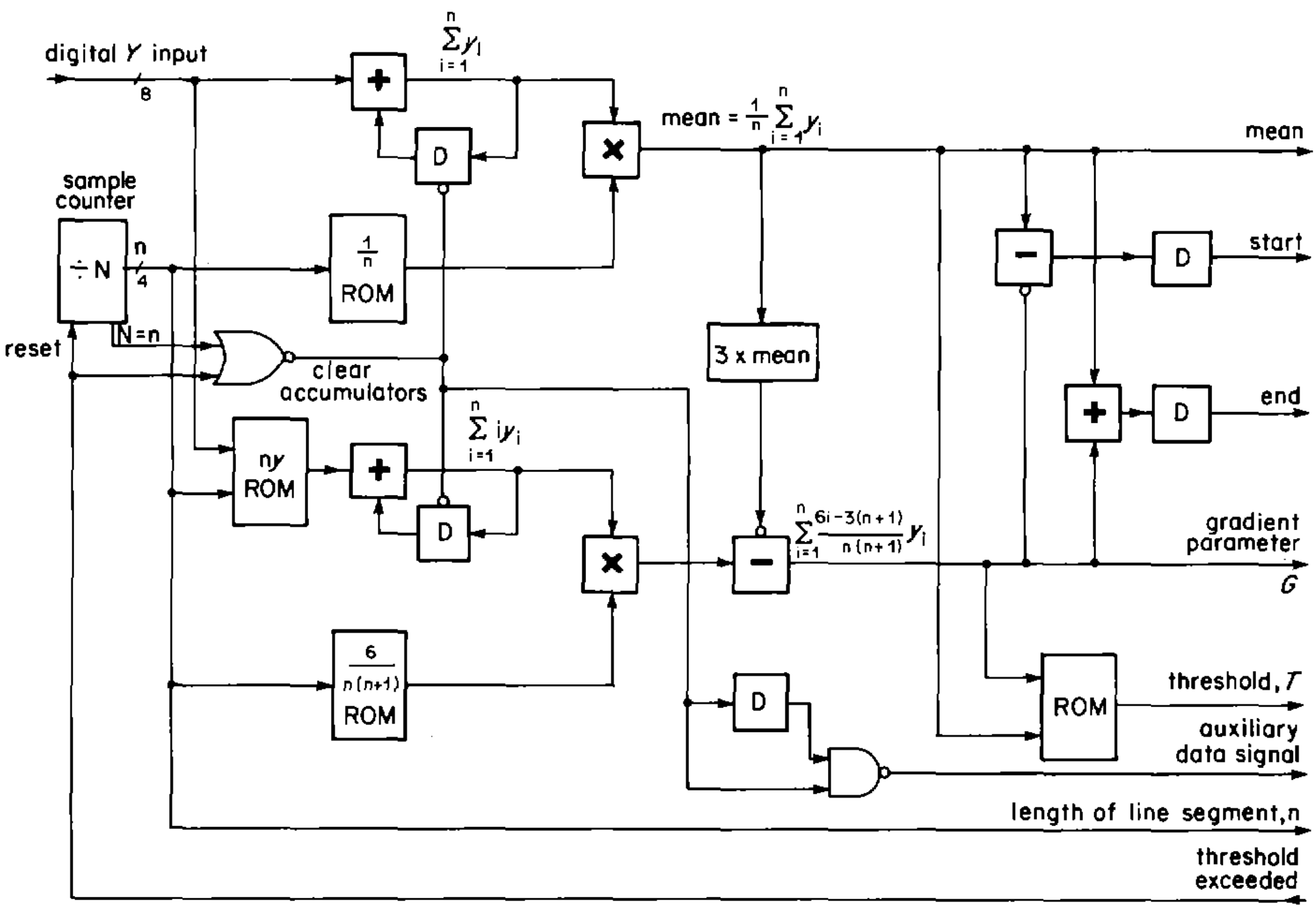


Fig. 6(b) - Block diagram of generator of best-fit slope parameters.

operated on a block-by-block basis. The activity measure for a block is an estimate of how the block has changed from the previous frame to the current frame. For example:

$$A(\mathrm{n}) = \sum_{\mathrm{i}} [x_{\mathrm{i}}(\mathrm{n}) - x'_{\mathrm{i}}(\mathrm{n}-1)]^2$$

where $x_{\mathrm{i}}(\mathrm{n})$ is the input signal in the current frame and $x'_{\mathrm{i}}(\mathrm{n}-1)$ is the decoded signal in the previous frame.

As before, active blocks are transmitted at high or full resolution, but inactive blocks are reconstructed using information from the previous frame. One way to do this is simply to repeat the block from the previous frame, as in a conditional replenishment system. However, it was found that better results could be obtained by transmitting one sample corresponding to the mean of the block in the current frame, and updating the 'shape' (i.e. the samples minus the mean) from the block in the previous frame:

$$x'_{\mathrm{i}}(\mathrm{n}) = \overline{x}(\mathrm{n}) + x'_{\mathrm{i}}(\mathrm{n}-1) - \overline{x}'(\mathrm{n}-1).$$

This enables a low-resolution version of the picture to be reconstructed immediately after, for example, a scene change.

A significant and novel improvement to the basic technique was achieved by using a cumulative version of the activity measure, as follows:

$$A_{\mathrm{c}}(\mathrm{n}) = \mathrm{k}.A_{\mathrm{c}}(\mathrm{n}-1) + \sum_{\mathrm{i}} [x_{\mathrm{i}}(\mathrm{n}) - x'_{\mathrm{i}}(\mathrm{n}-1)]^2$$

where k is some 'decay' constant (a typical value being 0.5) and the accumulating sum $A_c(\mathrm{n})$ is cleared whenever the block is transmitted at high resolution. This refinement ensures that all blocks are updated within a few frames.

Using this technique, and making the assumptions given in Section 1, it is possible with 4 x 4 blocks to transmit an average proportion $p = 0.20$ of the blocks at high resolution. The picture quality obtainable at this value of p is usable, but rather noisy and probably unacceptable for normal broadcast transmission. One way to increase the value of p significantly is to transmit the active blocks at half the full sample rate, by subsampling the block, for example, in a field quincunx pattern. With this technique, nearly half the blocks can be transmitted at high resolution and the subjective quality is good; however, because of the filtering and interpolation required for the field quincunx subsampling, the full range of spatial frequencies that ought to be carried in an HDTV system is not available and impairments can be seen on some critical moving sequences.

4.4 Motion compensated BASS coding

It is possible to improve the picture quality by associating, with each inactive block, a motion vector which is used to improve the reconstruction of these inactive blocks from the previous frame information. In order to limit the additional data capacity that would be required to signal the motion vectors it is necessary to increase the block size to, say, 8 x 8 sample points. Some experiments involving motion compensated BASS coding have been carried out and these are described below.

The average motion for each 8 x 8 block was measured by an exhaustive search in which motion vectors extending to ± 7 pixels horizontally and ± 7 field lines vertically were applied in turn to the block. For each motion vector, the resulting mean square difference with the corresponding block in the previous decoded frame was measured, the chosen vector being the one that yielded the smallest such difference. Vertical motion was measured only to the nearest field line (rather than picture line) because of the problems associated with interlace. The motion measurement process resulted in a motion vector which could be described by an 8-bit number and transmitted as part of the digital data signal. More sophisticated motion measurement techniques could also be used, for example, a method which calculates motion vectors to sub-pixel accuracy[9].

Each inactive block was reconstructed using the block in the previous frame displaced according to the motion vector as illustrated in Fig. 7. In order to calculate a suitable activity measure the temporal activity measure was first modified to include the motion estimate, i.e.

$$A(\mathrm{n}) = \sum_{i,j} [x(\mathrm{i,j,n}) - x(\mathrm{i} - d_x(\mathrm{i,j,n}), \mathrm{j} - d_y(\mathrm{i,j,n}), \mathrm{n} - 1)]^2$$

where $d_x(\mathrm{i,j,n})$ and $d_y(\mathrm{i,j,n})$ are the horizontal and vertical motion vector measurements for a block in the current frame. A cumulative measure, which approximated accumulation along the direction of motion, was then defined in the following manner. An accumulating partial activity measure $A_{\mathrm{cp}}(\mathrm{i,j,n} - 1)$ was first assigned to each pixel in the previous frame by sharing out the accumulating measure of a given block between the samples of that block.

i.e. $$A_{\mathrm{cp}}(\mathrm{i,j,n} - 1) = \frac{A_c(\mathrm{n} - 1)}{\mathrm{N_b}}$$

where N_b is the number of pixels per block. Then for

a given block in the current frame

$$A_c(n) = k \sum_{i,j} A_{cp}(i - d_x, j - d_y, n - 1) + A(n)$$

where the summation is over all i and j within the block in the current frame. Again, a given cumulative sum is cleared when it exceeds the threshold value and that block is sent to high resolution. This version of potential for better decoded picture quality than can be achieved with slope coding. However, both the coder and the decoder are likely to be more complicated than for slope coding.

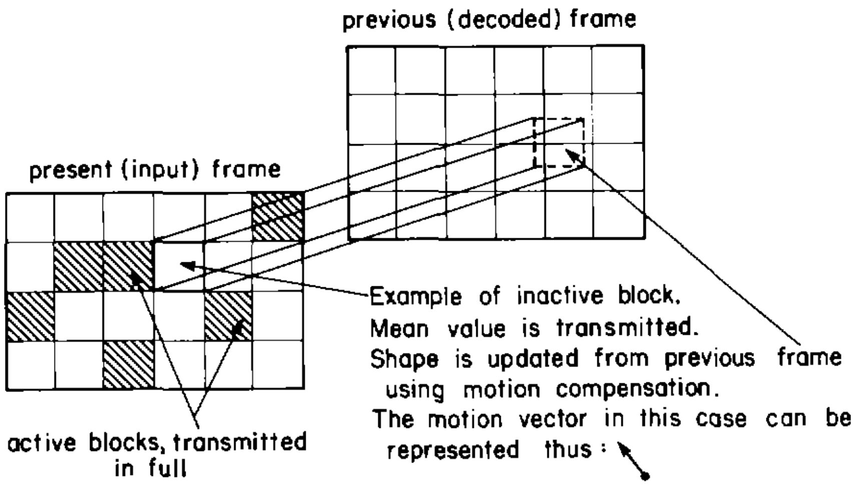


Fig. 7 - Principle of motion compensated BASS coding.

the motion-compensated cumulative activity measure works well in areas of the picture where there is global motion of the image as, for example, with a camera pan.

The data signal now consists of one bit per block of signalling together with an 8-bit motion vector for each inactive block. Assuming that the analogue component of the signal consists of all the samples of each active block and one sample for each inactive block (corresponding to the mean of that block), then the proportion of blocks that can be sent to full resolution is now $p = 0.26$. An example of the picture quality obtained from such a system is given in Fig. 8, together with an indication of the positions of active blocks. These positions may appear somewhat arbitrary in relatively inactive areas of the picture; this is accounted for by the cumulative activity measure which is ensuring that such blocks are updated on a rolling basis. The decoded picture appears to be virtually unimpaired; however, it should be noted that with a moving sequence the decoded picture appears slightly noisier than the source picture.

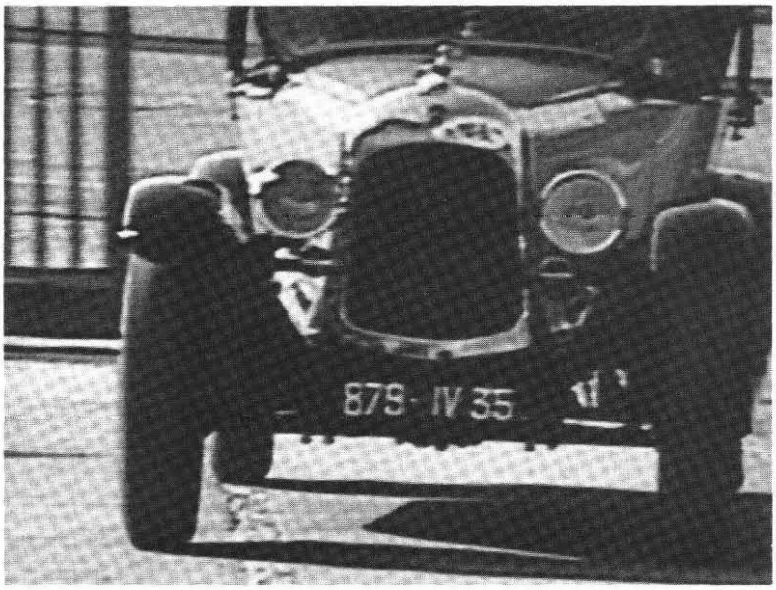
BASS coding, with motion compensation and a cumulative activity measure, appears to have

4.5 Practical considerations

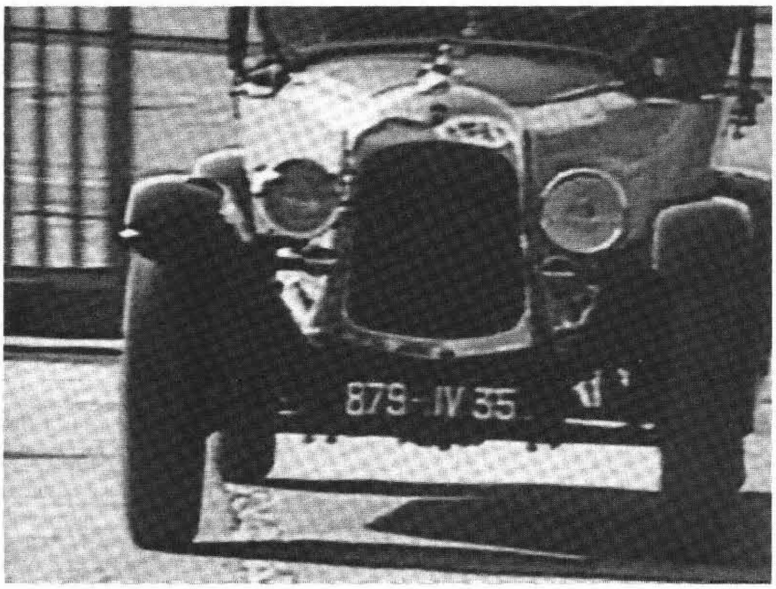
A block diagram of a possible hardware implementation of a BASS coder is given in Fig. 9. This particular implementation updates the whole of each inactive block, rather than just its shape, from the previous field. It also uses a 'two-pass' approach, in which a fixed number of active blocks per frame are transmitted. In the first pass, the activity measures are calculated for all the blocks in the frame. The threshold is then calculated and used in the second pass, in which the BASS coding itself is carried out.

The two-pass approach has some advantages over the more common 'one-pass' approach involving buffer feedback control. One advantage is that the number of transmitted samples and digital data bits corresponding to each frame is constant; this aids recovery of frame synchronization following a synchronization error, or at startup. Another advantage is that the two-pass system responds more quickly to changes in overall picture activity; for example, the inevitable reduction in picture quality at a scene change will be more confined to the frame immediately after the scene change, where it is less visible than usual. However, the 'two-pass' approach leads to greater complexity in the decoder; in particular, a second full frame store is required as shown in Fig. 9.

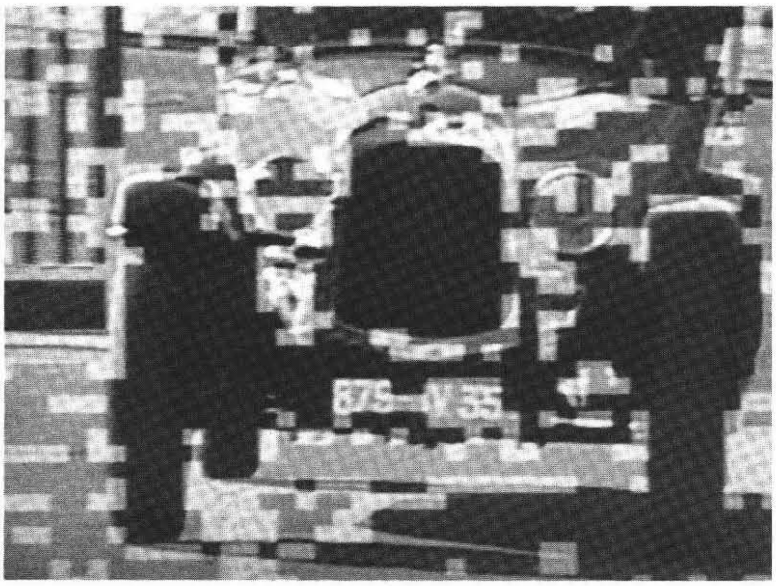
Fig. 8 - BASS coding results.



(a) One frame of the 'Voiture' source sequence.



(b) Picture after BASS coding and reconstruction.



(c) BASS-coded picture with 'active' blocks highlighted.

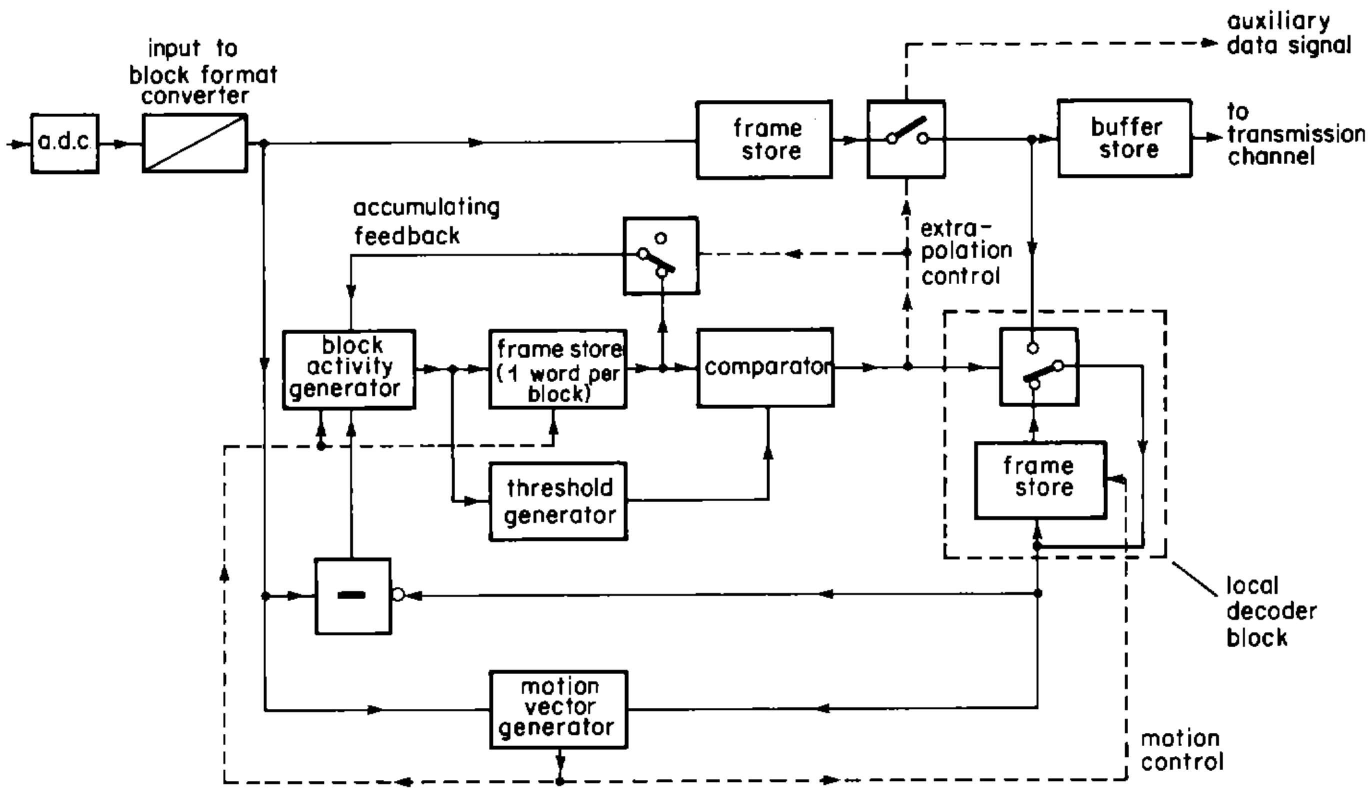


Fig. 9 - Block diagram of BASS coder.

5. CONCLUSIONS

Two possible methods using digital assistance for transmitting HDTV signals in a DBS channel have been described and the results of preliminary studies given. The two methods, known as slope coding and BASS coding, show promise as approaches to a 'non-compatible' HDTV coding standard. Both techniques lead to simple decoders and relatively complex coders. BASS coding appears to offer better picture quality than slope coding, but at higher overall complexity.

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APPENDIX 1

A Lower Bound on the Signalling Rate for Slope Coding

Consider a slope coding data signal for the system of Section 3.2 in which picture samples are transmitted in pairs. The information contained in the data signal is equivalent to an imaginary data signal which takes the value **0** for every sample in the original picture signal that is not transmitted and **1** for every pair of samples that is transmitted. We assume that there is no correlation between successive values of this data signal.

For a large number, N say, of input picture samples, there will be on average $N(1-p)$ samples that are not transmitted and $Np/2$ pairs of transmitted samples. There will therefore be a total of $N(1-p/2)$ symbols in the new data signal, of which a proportion $q = (p/2)/(1-p/2)$ will take the value **1** and a proportion $1-q$ will take the value **0**.

A lower bound on the signalling rate needed to describe the new data signal is the information content or entropy of that signal, which in this case is $H' = -q\log_2 q - (1-q)\log_2(1-q)$ bits per symbol. Measured in bits per input picture sample, this quantity is

$$H = (1-p/2) \,.\, H' = -(1-p/2) \,.\, [q \log_2 q + (1-q)\log_2(1-q)].$$

APPENDIX 2

Recursive Algorithm for Best-fit Slope Parameter Generation

First, we derive the parameters A and B for the equation of a best-fit straight line through n input samples $y_1,\ldots,y_n$:

$$y = A[i - ½(n+1)] + B + Y$$

where $Y = \sum_1^n y_i$, the mean of the input samples.

We require to find A and B to minimize E, the mean square error between the straight line and the input samples:

$$E = \frac{1}{n} \sum_1^n \{ A[i - ½(n+1)] + B + Y - y_i \}^2 \cdot$$

To minimize E, we obtain the following pair of equations:

$$\frac{\partial E}{\partial A} = 2 \sum_1^n \{ A[i - ½(n+1)] + B + Y - y_i \} \cdot [i - ½(n+1)] = 0$$

$$\frac{\partial E}{\partial B} = 2 \sum_1^n \{ A[i - ½(n+1)] + B + Y - y_i \} \qquad = 0$$

which give

$$A = \frac{\sum_{1}^{n} i(y_i - Y)}{\sum_{1}^{n} [i - ½(n+1)]^2}$$

$$B = 0 \cdot$$

Simplifying the expression for A:

$$A = \sum_{1}^{n} \frac{[12i - 6(n+1)]y_i}{n(n+1)(n-1)} .$$

It would be possible for the generator of best-fit slope parameters to calculate the gradient A directly according to the above formula. However, A would have to be calculated and stored either as a floating-point number or as a fixed-point number requiring many bits. This is because A has to be specified to high accuracy for larger values of n, whereas for smaller values of n a large range is needed. To avoid this problem, a 'gradient parameter' G is used instead:

$$G = A \,.\, ½(n-1) = \sum_{1}^{n} \frac{[6i - 3(n+1)]y_i}{n(n+1)} .$$

G is equal to half the 'height' of the interpolated line segment. It therefore has the approximate dimensions of sample values themselves.

The equation of the best-fit straight line is therefore

$$y = G \,.\, C_i(n) + Y$$

where $C_i(n) = \dfrac{2i - n - 1}{n-1}$.

Fig. 6(b) shows how G is calculated recursively using accumulated sums of y_i and of iy_i, and lookup tables containing $1/n$ and $6/[n(n + 1)]$.

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